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Please find below and/or attached an Office communication concerning this application or proceeding.

The time period for reply, if any, is set in the attached communication.

Office Action Summary	Application No.	Applicant(s)	
	10/519,000	CHRISTENSEN ET AL.	
	Examiner	Art Unit	
	MICHAEL C. COLUCCI	2626	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE *3* MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on *08 April 2009*.

2a)☒ This action is **FINAL**. 2b)☐ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) *1-24* is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) *1-24* is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☐ The drawing(s) filed on _____ is/are: a)☐ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☐ Notice of References Cited (PTO-892)
2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)
3) ☒ Information Disclosure Statement(s) (PTO/SB/08) Paper No(s)/Mail Date _____.
4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .
5) ☐ Notice of Informal Patent Application
6) ☐ Other: _____.

DETAILED ACTION

Response to Arguments

1. Applicant's arguments filed 04/08/2009 have been fully considered but they are not persuasive.

NOTE: Examiner would like to remind Applicant of the following:

"USPTO personnel are to give claims their broadest reasonable interpretation in light of the supporting disclosure. In re Morris, 127 F.3d 1048, 1054-55, 44 USPQ2d 1023,1027-28 (Fed. Cir. 1997). Limitations appearing in the specification but not recited in the claim should not be read into the claim. E-Pass Techs., Inc. v. 3Com Corp., 343 F.3d1364, 1369, 67 USPQ2d 1947, 1950 (Fed. Cir. 2003) (claims must be interpreted "in view of the specification" without importing limitations from the specification into the claims unnecessarily). In re Prater, 415 F.2d 1393, 1404-05, 162 USPQ 541, 550-551 (CCPA 1969). See also In re Zletz, 893 F.2d 319, 321-22, 13 USPQ2d 1320, 1322 (Fed. Cir. 1989) ("During patent examination the pending claims must be interpreted as broadly as their terms reasonably allow.... The reason is simply that during patent prosecution when claims can be amended, ambiguities should be recognized, scope and breadth of language explored, and clarification imposed.... An essential purpose of patent examination is to fashion claims that are precise, clear, correct, and unambiguous. Only in this way can uncertainties of claim

scope be removed, as much as possible, during the administrative process."). Where an explicit definition is provided by the applicant for a term, that definition will control interpretation of the term as it is used in the claim. Toro Co. v. White Consolidated Industries Inc., 199 F.3d 1295, 1301, 53 USPQ2d 1065, 1069 (Fed. Cir. 1999) (meaning of words used in a claim is not construed in a "lexicographic vacuum, but in the context of the specification and drawings."). Any special meaning assigned to a term "must be sufficiently clear in the specification that any departure from common usage would be so understood by a person of experience in the field of the invention." Multiform Desiccants Inc. v. Medzam Ltd., 133 F.3d 1473, 1477, 45 USPQ2d 1429, 1432 (Fed. Cir. 1998). See also MPEP § 2111.01."

Argument 1 (page 10 paragraph 1 & page 11 paragraph 3):

- "Therefore Adams does not disclose or suggest that a first measurement for an estimated bit time is the time separating a first set of successive identified transitions"
- "Fletcher fails to cure the deficiencies of Adams in this respect. Fletcher's technique for estimating bit times does not rely on bit windows, but instead builds a bit time estimate by progressively increasing the average value from an initial value of zero. Because Fletcher takes an alternate approach for estimating bit times, Fletcher has no need for bit window times. Therefore, applicants respectfully assert that Fletcher does not disclose or

suggest estimating minimum or maximum bit window times. As above, Smyth does nothing to cure the deficiencies of Adams and Fletcher. The Smyth patent does not concern itself with biphase encoding, and as such never deals with bit window times at all. Therefore, Adams, Fletcher, and/or Smyth, taken alone or in any combination, fail to disclose or suggest estimating minimum or maximum bit window times"

Response to argument 1:

Adams is directly within the scope of the present invention, wherein Adams like the present invention teaches bi-phase decoding with scaled bit times between .5 and 1.5. Further, "the time separating a first set of successive identified transitions is a first measurement of said estimated bit time" is described in the specification, wherein two successive subframes merely represent a frame (present invention spec. page 13 par. 2). Further, "the time separating a first set of successive identified transitions" is broadly described, wherein the times between these transitions is analyzed to find the short and long times between (i.e. .5 - 1.5) (present invention spec. page 11 par. 3).

Furthermore, consider figures 5 and 6 of the present invention that disclose a depiction of sub frames and frames containing preambles having varying time segments relative to biphase decoding with AES standards, wherein bit times vary based on the preamble type as depicted in Table 1 (present invention spec. Table 1 & Fig. 5-6). Adams teaches AES biphase decoding having the same well known analysis performed, wherein Adams teaches a

preamble, subframes, and frames in succession separated by a varying bit time (page 8 Table and Fig. 2-4). Adams also explicitly teaches the transition between boundaries of frames in a signal like the present invention, wherein x, y, and z preambles are explicitly taught having varying durations (page 3 lines 26-31).

Additionally, Adams explicitly teaches a period of time detected between transitions within a frame of AES biphase data, wherein this time is the time of incoming bits i.e. bit time (page 8 lines 1-21). Thus Adams teaches time measurement within a frame ranging from a minimum ½ to a maximum 1.5.

Though Adams clearly teaches largest and smallest bit times in a subframe/frame transition, Smyth has been incorporated to further teach obvious limitations that Adams does not disclose, such as the use of largest and smallest bit window times. Further, in view of the specification, a window is not understood to be used in the traditional sense as function, instead it is merely a semantic variation of a portion or segment taken, similar to a frame or subframe of a signal and is therefore functionally equivalent to a frame, i.e. bit time within a frame with large and small times. A window is merely a segment from a data stream which contains the data of interest. Smyth explicitly teaches the use of windows, wherein Smyth teaches that window size is selected as a function of the ratio of the transmission rate to the encoder sampling rate so that the size of the output frame is constrained to lie in a desired range. When the amount of compression is relatively low the window size is reduced so that the frame size

does not exceed an upper maximum. As a result, a decoder can use an input buffer with a fixed and relatively small amount of RAM. When the amount of compression is relatively high, the window size is increased. As a result, the GBM system can distribute bits over a larger time window thereby improving encoder performance (Smyth Col. 3 line 65 – Col. 4 line 10).

Further, Smyth teaches the amount of RAM required at the decoder to buffer the incoming data stream is kept relatively low, which reduces the cost of the decoder. At low rates larger window sizes can be used to frame the PCM data, which improves the coding performance. At higher bit rates, smaller window sizes must be used to satisfy the data constraint. This necessarily reduces coding performance, but at the higher rates it is insignificant. Also, the manner in which the PCM data is framed allows the decoder 18 to initiate playback before the entire output frame is read into the buffer. This reduces the delay or latency of the audio coder (Smyth Col. 6 lines 1-12).

The teachings of Smyth are consistent with Adams as well as the present invention, wherein the ambiguity of the conflicting terminology of short/long (present invention spec. page 11) is understood as a short/long bit time in correlation with a short/long duration. Though, Adams reads upon identifying time between transitions, Examiner has described teachings from Smyth to clarify window/frame teachings. See rejection.

Argument 2 (page 10 paragraph 5):

- “Furthermore, claim 23 recites, inter alia, "estimating minimum and maximum bit window times." Claim 9, 19, and 24 recite analogous language. The Examiner asserts that the Adams patent discloses this element in the form of the shift register shown in FIG. 11. However, the basis for the Examiner's assertion remains unclear. Adam's shift register, serves to discriminate between different pulse lengths. Adams makes use of delay taps, with the delay length based on a continually adjusting servo loop, whereas the applicants' claimed invention constructs a timing window based on an estimated bit time. In order to estimate a bit time applicants estimate a bit window as described in the present application, pages 11 and 12. The bit window and timing window have ranges which represent different quantities. Adams does not have any analogue for bit window times”

Response to argument 2:

Additionally, Adams explicitly teaches a period of time detected between transitions within a frame of AES biphase data, wherein this time is the time of incoming bits i.e. bit time (page 8 lines 1-21). Thus Adams teaches time measurement within a frame ranging from a minimum ½ to a maximum 1.5.

Though Adams clearly teaches largest and smallest bit times in a subframe/frame transition, Smyth has been incorporated to further teach obvious limitations that Adams does not disclose, such as the use of largest and smallest

bit window times. Further, in view of the specification, a window is not understood to be used in the traditional sense as function, instead it is merely a semantic variation of a portion or segment taken, similar to a frame or subframe of a signal and is therefore functionally equivalent to a frame, i.e. bit time within a frame with large and small times. A window is merely a segment from a data stream which contains the data of interest. Smyth explicitly teaches the use of windows, wherein Smyth teaches that window size is selected as a function of the ratio of the transmission rate to the encoder sampling rate so that the size of the output frame is constrained to lie in a desired range. When the amount of compression is relatively low the window size is reduced so that the frame size does not exceed an upper maximum. As a result, a decoder can use an input buffer with a fixed and relatively small amount of RAM. When the amount of compression is relatively high, the window size is increased. As a result, the GBM system can distribute bits over a larger time window thereby improving encoder performance (Smyth Col. 3 line 65 – Col. 4 line 10).

Further, Smyth teaches the amount of RAM required at the decoder to buffer the incoming data stream is kept relatively low, which reduces the cost of the decoder. At low rates larger window sizes can be used to frame the PCM data, which improves the coding performance. At higher bit rates, smaller window sizes must be used to satisfy the data constraint. This necessarily reduces coding performance, but at the higher rates it is insignificant. Also, the manner in which the PCM data is framed allows the decoder 18 to initiate

playback before the entire output frame is read into the buffer. This reduces the delay or latency of the audio coder (Smyth Col. 6 lines 1-12).

The teachings of Smyth are consistent with Adams as well as the present invention, wherein the ambiguity of the conflicting terminology of short/long (present invention spec. page 11) is understood as a short/long bit time in correlation with a short/long duration. Though, Adams reads upon identifying time between transitions, Examiner has described teachings from Smyth to clarify window/frame teachings. See rejection.

Claim Rejections - 35 USC § 103

1. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

2. Claims 1-24 are rejected under 35 U.S.C. 103(a) as being unpatentable over Adams WO9816040 (Hereinafter Adams) in view Smyth et al. US 5956674 A (Hereinafter Smyth) and further in view of Fletcher et al. EP0453063 (Hereinafter Fletcher).

Re claim 1, Adams teaches a method of extracting digital audio data words from a serialized stream of digital audio data (page 1 lines 22-27), comprising:

constructing a timing window from an estimated bit time for said serialized stream of digital audio data, said timing window having a preamble sub-window and at least one data sub-window (page 3 lines 1-18);

extracting plural digital audio data words from said serialized stream of digital audio based upon the location of each transition in said serialized stream of digital audio data (page 1 lines 22-27) relative to said preamble sub-window and said at least one data sub-window of said timing window (page 3 lines 1-18),

each one of said extracted plural digital audio data word having a preamble identifiable by a combination of at least one transition located in said preamble sub-window of said timing window and at least one transition located in said at least one data sub- window of said timing window (page 3 lines 1-18)

the time separating a first set of successive identified transitions is a first measurement of said estimated bit time (page 3 lines 26-31)

However, Adams fails to teach extracted plural digital audio data words

Smyth teaches multiple channels of PCM audio data 14, typically sampled at 48 kHz with word lengths between 16 and 24 bits, into a data stream 16 at a known transmission rate, suitably in the range of 32-4096 kbps. Unlike known audio coders, the present architecture can be expanded to higher sampling rates (48-192 kHz) without making the existing decoders, which were designed for the baseband sampling rate or any intermediate sampling rate, incompatible. Furthermore, the PCM data 14 is windowed and encoded a frame at a time where each frame is preferably split into 1-4 subframes. The size of the audio window, i.e. the number of PCM samples, is based on

the relative values of the sampling rate and transmission rate such that the size of an output frame, i.e. the number of bytes, read out by the decoder 18 per frame is constrained, suitably between 5.3 and 8 kbytes (Smyth Col. 5 lines 52-67).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to modify the system of Adams to incorporate extracted plural digital audio data words as taught by Smyth to allow for an increased sampling rate for frames and subframes on a compatible level with the extracted audio signal (Smyth Col. 5 lines 52-67).

However, Adams in view of Smyth fails to teach wherein said bit time is estimated by averaging a plurality of data stream pulse lengths.

Fletcher teaches the average value register 28 holds an average value of the short pulse length, as measured by circuit 14. It is convenient to call the short pulses A pulses, and the long pulses B pulses. The average is thus calculated by halving the measured length of the B pulses and then taking a time average of the A and 1/2 B pulse lengths. The term average is here used in a general rather than a strict mathematical sense and the precise function can be chosen in many ways to produce a running average or mean value derived from the recently received input values as will be apparent to those skilled in the art (Fletcher Col. 3 lines 7-20).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to modify the system of Adams in view of Smyth to incorporate wherein said bit time is estimated by averaging a plurality of data stream pulse lengths

as taught by Fletcher to allow for well known method to produce precision averaging of audio frames based on pulse length (Fletcher Col. 3 lines 7-20).

Re claims 2, 15, and 21, Adams teaches the method of claim 1, and further comprising identifying said extracted data words as having a first type of preamble of said extracted data words have a pair of successive transitions located in said preamble sub-window followed by a pair of successive transitions located in said at least one data sub-window (page 3 lines 1-18 & Fig. 11).

Re claims 3 and 16, Adams teaches the method of claim 2, and further comprising identifying said extracted data words as having a second type of preamble if said extracted dam words have a pair of non-successive transitions located in said preamble sub-window-separated by a pair of successive transitions located in said at least one data sub-window (page 3 lines 1-18 & Fig. 11).

Re claims 4 and 17, Adams teaches the method of claim 3, and further comprising identifying said extracted data words as having a third type of preamble if said extracted data words have a transition located in said preamble sub-window followed by first, second and third transitions located in said at least one data sub-window (page 3 lines 1-18).

Re claim 5, Adams teaches the method of claim 4, wherein said timing window is constructed such that said preamble sub-window extends from about 1¼ times said estimated bit time to about 1¾ times said estimated bit time (page 8 lines 8-21).

Re claim 6, Adams teaches the method of claim 5, wherein said timing window is constructed such that said at least one data sub-window extends from about ¼ times said estimated bit time to about 1¼ times said estimated bit time (page 8 lines 8-21).

Re claim 7, Adams teaches the method of claim 4, wherein said timing window is constructed such that said at least one data sub-window includes a first data sub-window which extends from about ¼ times said estimated bit time to about 3/4 times said estimated bit time and a second data sub-window which extends from about 3/4 times said estimated bit time to about 1¼ times said estimated bit time (page 8 lines 8-21).

Re claims 8 and 18, Adams teaches the method of claim 1, wherein said estimated bit time is derived from said serialized stream of digital audio data (page 1 lines 22-27).

Re claims 9 and 19, Adams teaches the method of claim 8, and further comprising: estimating minimum and maximum bit window times (page 8 lines 8-21 &

fig. 11) constructing a bit window from said minimum and maximum bit window times (page 6 lines 20-27);

identifying transitions in said serialized stream of digital audio data (page 1 lines 22-27) which occur within said constructed bit window (page 1 lines 22-27),

Though necessary and understood as frames of interest, Adams does not discretely teach window times

Smyth teaches that window size is selected as a function of the ratio of the transmission rate to the encoder sampling rate so that the size of the output frame is constrained to lie in a desired range. When the amount of compression is relatively low the window size is reduced so that the frame size does not exceed an upper maximum. As a result, a decoder can use an input buffer with a fixed and relatively small amount of RAM. When the amount of compression is relatively high, the window size is increased. As a result, the GBM system can distribute bits over a larger time window thereby improving encoder performance (Smyth Col. 3 line 65 – Col. 4 line 10).

Further, Smyth teaches the amount of RAM required at the decoder to buffer the incoming data stream is kept relatively low, which reduces the cost of the decoder. At low rates larger window sizes can be used to frame the PCM data, which improves the coding performance. At higher bit rates, smaller window sizes must be used to satisfy the data constraint. This necessarily reduces coding performance, but at the higher rates it is insignificant. Also, the manner in which the PCM data is framed allows the decoder 18 to initiate playback before the entire output frame is read into the buffer. This reduces the delay or latency of the audio coder (Smyth Col. 6 lines 1-12)

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to modify the system of Adams to incorporate constructing a bit window from said minimum and maximum bit window times as taught by Smyth to allow for an increased sampling rate for frames and subframes on a compatible level with the extracted audio signal (Smyth Col. 5 lines 52-67), wherein compatibility allows for an appropriate allocation of memory within a window, wherein a window size is variable to allow for higher encoding performance (Smyth Col. 3 line 65 – Col. 4 line 10).

Re claim 10, Adams in view of Smyth fails to teach the method of claim 9, further comprising determining said estimated bit time from a running average of plural measurements of said estimated bit time.

Fletcher teaches the average value register 28 holds an average value of the short pulse length, as measured by circuit 14. It is convenient to call the short pulses A pulses, and the long pulses B pulses. The average is thus calculated by halving the measured length of the B pulses and then taking a time average of the A and 1/2 B pulse lengths. The term average is here used in a general rather than a strict mathematical sense and the precise function can be chosen in many ways to produce a running average or mean value derived from the recently received input values as will be apparent to those skilled in the art (Fletcher Col. 3 lines 7-20).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to modify the system of Adams in view of Smyth to incorporate

determining said estimated bit time from a running average of plural measurements of said estimated bit time as taught by Fletcher to allow for well known method to produce precision averaging of audio frames based on pulse length (Fletcher Col. 3 lines 7-20).

Re claim 11, Adams teaches a method of extracting digital audio data words from a serialized stream of digital audio data (page 1 lines 22-27), comprising:

constructing a timing window from an estimated bit time for said serialized stream of digital audio data, said timing window having a preamble sub-window and at least one data sub-window (page 3 lines 1-18);

extracting plural digital audio data words from said serialized stream of digital audio based upon the location of each transition in said sampled stream of digital audio data (page 1 lines 22-27) relative to said preamble sub-window and said at least one data sub-window of said timing window wherein said bit time is estimated by averaging a plurality of data stream pulses (page 3 lines 1-18), the time separating a first set of successive identified transitions is a first measurement of said estimated bit time (page 3 lines 26-31).

However, Adams in view of Smyth fails to teach sampling said serialized stream of digital audio data at a fast sample rate

Fletcher teaches that to be able to detect the pulses in a real situation with data sample rates of up to 54 KHz and bit rates of 64 times that requires a high sampling rate to be used in the circuit 14. Measurement sampling rates of at least six or seven times the maximum input pulse rate are desirable. The high frequency clock signal required

may be derived externally using a conventional crystal-based oscillator (Fletcher Col. 6 lines 4-11).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to modify the system of Adams in view of Smyth to incorporate sampling said serialized stream of digital audio data at a fast sample rate as taught by Fletcher to allow for the detection of live pulses at a high sampling rate (Fletcher Col. 6 lines 4-11).

Re claim 12, Adams in view of Smyth fails to teach the method of claim 11, wherein said fast sample rate is at least about twenty times faster than a data rate for said serialized stream of digital audio data.

Fletcher teaches that to be able to detect the pulses in a real situation with data sample rates of up to 54 KHz and bit rates of 64 times that requires a high sampling rate to be used in the circuit 14. Measurement sampling rates of at least six or seven times the maximum input pulse rate are desirable. The high frequency clock signal required may be derived externally using a conventional crystal-based oscillator (Fletcher Col. 6 lines 4-11).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to modify the system of Adams in view of Smyth to incorporate fast sample rate is at least about twenty times faster than a data rate for said serialized stream of digital audio data as taught by Fletcher to allow for the detection of live pulses at a high sampling rate (Fletcher Col. 6 lines 4-11).

Re claim 13, Adams in view of Smyth fails to teach the method of claim 12, wherein said fast sample rate is derived from a fast clock having a frequency of at least about twenty times faster than the frequency of said serialized stream of digital data.

Fletcher teaches that to be able to detect the pulses in a real situation with data sample rates of up to 54 KHz and bit rates of 64 times that requires a high sampling rate to be used in the circuit 14. Measurement sampling rates of at least six or seven times the maximum input pulse rate are desirable. The high frequency clock signal required may be derived externally using a conventional crystal-based oscillator (Fletcher Col. 6 lines 4-11).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to modify the system of Adams in view of Smyth to incorporate fast sample rate is at least about twenty times faster than a data rate for said serialized stream of digital audio data as taught by Fletcher to allow for the detection of live pulses at a high sampling rate (Fletcher Col. 6 lines 4-11).

Re claim 14, Adams teaches the method of claim 13, wherein each one of said extracted plural digital audio data words has a preamble identifiable by a combination of at least one transition located in said preamble sub-window of said timing window and at least one transition located in said at least one data sub-window of said timing window (page 3 lines 1-18).

However, Adams fails to teach extracted plural digital audio data words

Smyth teaches multiple channels of PCM audio data 14, typically sampled at 48 kHz with word lengths between 16 and 24 bits, into a data stream 16 at a known transmission rate, suitably in the range of 32-4096 kbps. Unlike known audio coders, the present architecture can be expanded to higher sampling rates (48-192 kHz) without making the existing decoders, which were designed for the baseband sampling rate or any intermediate sampling rate, incompatible. Furthermore, the PCM data 14 is windowed and encoded a frame at a time where each frame is preferably split into 1-4 subframes. The size of the audio window, i.e. the number of PCM samples, is based on the relative values of the sampling rate and transmission rate such that the size of an output frame, i.e. the number of bytes, read out by the decoder 18 per frame is constrained, suitably between 5.3 and 8 kbytes (Smyth Col. 5 lines 52-67).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to modify the system of Adams to incorporate extracted plural digital audio data words as taught by Smyth to allow for an increased sampling rate for frames and subframes on a compatible level with the extracted audio signal (Smyth Col. 5 lines 52-67).

Re claim 20, Adams teaches a bi-phase decoder for use in decoding a stream of AES-3 digital audio data (page 1 lines 22-27), comprising:

a decoder circuit coupled to receive a stream of AES-3 digital audio data, an estimated bit time for said stream of AES-3 digital audio data (page 1 lines 22-27) and a

fast clock, said fast clock having a frequency of about at least twenty times faster than a frequency of said stream of AES-3 digital audio data; and

a data store coupled to said decoder circuit, 'said data store receiving subframes of digital audio data extracted, from said stream of AES-3 digital audio data by said decoder circuit (page 3 lines 1-18);

said decoder circuit extracting subframes of said digital audio data by constructing a timing window from said estimated bit time, sampling said stream of AES-3 digital audio data using said fast clock and applying said sampled stream of AES-3 digital audio data to said timing window to identify transitions, in said sampled stream of AES-3 digital audio data, indicative of preambles of said subframes of digital audio data (page 3 lines 1-18)

the time separating a first set of successive identified transitions is a first measurement of said estimated bit time (page 3 lines 26-31)

However, Adams in view of Smyth fails to teach said bit time is estimated by averaging a plurality of data stream pulses

a fast clock having a frequency of about at least twenty times faster than a frequency of said stream of AES-3 digital audio data

Fletcher teaches that to be able to detect the pulses in a real situation with data sample rates of up to 54 KHz and bit rates of 64 times that requires a high sampling rate to be used in the circuit 14. Measurement sampling rates of at least six or seven times the maximum input pulse rate are desirable. The high frequency clock signal required

may be derived externally using a conventional crystal-based oscillator (Fletcher Col. 6 lines 4-11).

Fletcher also teaches the average value register 28 holds an average value of the short pulse length, as measured by circuit 14. It is convenient to call the short pulses A pulses, and the long pulses B pulses. The average is thus calculated by halving the measured length of the B pulses and then taking a time average of the A and 1/2 B pulse lengths. The term average is here used in a general rather than a strict mathematical sense and the precise function can be chosen in many ways to produce a running average or mean value derived from the recently received input values as will be apparent to those skilled in the art (Fletcher Col. 3 lines 7-20).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to modify the system of Adams in view of Smyth to incorporate bit time is estimated by averaging a plurality of data stream pulses and a fast clock having a frequency of about at least twenty times faster than a frequency of said stream of AES-3 digital audio data as taught by Fletcher to allow for well known method to produce precision averaging of audio frames based on pulse length (Fletcher Col. 3 lines 7-20) and for the detection of live pulses at a high sampling rate (Fletcher Col. 6 lines 4-11).

Re claim 22, Adams teaches the apparatus of claim 21, and further comprising a bit time estimator circuit having an input coupled to receive said stream (page 1 lines 22-27) of AES-3 digital audio data and an output coupled to said decoder circuit, said bit

time estimator determining said estimated bit time for output to said decoder circuit (Col. 1 lines 10-27).

Re claims 23 and 24, Adams teaches a method of extracting digital audio data words from a serialized stream of digital audio data (page 1 lines 22-27), comprising:

constructing a timing window from an estimated bit time for said serialized stream of digital audio data, said timing window having a preamble sub-window and at least one data sub-window (page 3 lines 1-18);

sampling said serialized stream of digital audio data (page 1 lines 22-27) at a fast sample rate; and extracting plural digital audio data words from said serialized stream of digital audio (page 1 lines 22-27) based upon the location of each transition in said sampled stream of digital audio data relative to said preamble sub-window and said at least one data sub-window of said timing window (page 3 lines 1-18);

estimating minimum (page 6 lines 20-27) and maximum bit window times (page 8 lines 8-21 & fig. 11);

constructing a bit window from said minimum and maximum bit window times; identifying transitions in said serialized stream of digital audio data which occur within said constructed bit window (page 3 lines 1-18), the time separating a set of successive identified transitions being a measurement of said estimated bit time (page 6 lines 20-27); and

However, Adams fails to teach extracted plural digital audio data words

window times

Smyth teaches multiple channels of PCM audio data 14, typically sampled at 48 kHz with word lengths between 16 and 24 bits, into a data stream 16 at a known transmission rate, suitably in the range of 32-4096 kbps. Unlike known audio coders, the present architecture can be expanded to higher sampling rates (48-192 kHz) without making the existing decoders, which were designed for the baseband sampling rate or any intermediate sampling rate, incompatible. Furthermore, the PCM data 14 is windowed and encoded a frame at a time where each frame is preferably split into 1-4 subframes. The size of the audio window, i.e. the number of PCM samples, is based on the relative values of the sampling rate and transmission rate such that the size of an output frame, i.e. the number of bytes, read out by the decoder 18 per frame is constrained, suitably between 5.3 and 8 kbytes (Smyth Col. 5 lines 52-67).

Smyth teaches that window size is selected as a function of the ratio of the transmission rate to the encoder sampling rate so that the size of the output frame is constrained to lie in a desired range. When the amount of compression is relatively low the window size is reduced so that the frame size does not exceed an upper maximum. As a result, a decoder can use an input buffer with a fixed and relatively small amount of RAM. When the amount of compression is relatively high, the window size is increased. As a result, the GBM system can distribute bits over a larger time window thereby improving encoder performance (Smyth Col. 3 line 65 – Col. 4 line 10).

Further, Smyth teaches the amount of RAM required at the decoder to buffer the incoming data stream is kept relatively low, which reduces the cost of the decoder. At

low rates larger window sizes can be used to frame the PCM data, which improves the coding performance. At higher bit rates, smaller window sizes must be used to satisfy the data constraint. This necessarily reduces coding performance, but at the higher rates it is insignificant. Also, the manner in which the PCM data is framed allows the decoder 18 to initiate playback before the entire output frame is read into the buffer. This reduces the delay or latency of the audio coder (Smyth Col. 6 lines 1-12)

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to modify the system of Adams to incorporate extracted plural digital audio data words and constructing a bit window from said minimum and maximum bit window times as taught by Smyth to allow for an increased sampling rate for frames and subframes on a compatible level with the extracted audio signal (Smyth Col. 5 lines 52-67), wherein compatibility allows for an appropriate allocation of memory within a window, wherein a window size is variable to allow for higher encoding performance (Smyth Col. 3 line 65 – Col. 4 line 10).

However, Adams in view of Smyth fails to teach a fast sample rate

determining said estimated bit time from a running average of plural measurements of said estimated bit time

Fletcher teaches that to be able to detect the pulses in a real situation with data sample rates of up to 54 KHz and bit rates of 64 times that requires a high sampling rate to be used in the circuit 14. Measurement sampling rates of at least six or seven times the maximum input pulse rate are desirable. The high frequency clock signal required

may be derived externally using a conventional crystal-based oscillator (Fletcher Col. 6 lines 4-11).

Fletcher also teaches the average value register 28 holds an average value of the short pulse length, as measured by circuit 14. It is convenient to call the short pulses A pulses, and the long pulses B pulses. The average is thus calculated by halving the measured length of the B pulses and then taking a time average of the A and 1/2 B pulse lengths. The term average is here used in a general rather than a strict mathematical sense and the precise function can be chosen in many ways to produce a running average or mean value derived from the recently received input values as will be apparent to those skilled in the art (Fletcher Col. 3 lines 7-20).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to modify the system of Adams in view of Smyth to incorporate a fast sample rate and determining said estimated bit time from a running average of plural measurements of said estimated bit time as taught by Fletcher to allow for well known method to produce precision averaging of audio frames based on pulse length (Fletcher Col. 3 lines 7-20) and for the detection of live pulses at a high sampling rate (Fletcher Col. 6 lines 4-11).

Conclusion

2. **THIS ACTION IS MADE FINAL.** Applicant is reminded of the extension of time policy as set forth in 37 CFR 1.136(a).

A shortened statutory period for reply to this final action is set to expire THREE MONTHS from the mailing date of this action. In the event a first reply is filed within TWO MONTHS of the mailing date of this final action and the advisory action is not mailed until after the end of the THREE-MONTH shortened statutory period, then the shortened statutory period will expire on the date the advisory action is mailed, and any extension fee pursuant to 37 CFR 1.136(a) will be calculated from the mailing date of the advisory action. In no event, however, will the statutory period for reply expire later than SIX MONTHS from the mailing date of this final action.

Any inquiry concerning this communication or earlier communications from the examiner should be directed to Michael C. Colucci whose telephone number is (571)-270-1847. The examiner can normally be reached on 9:30 am - 6:00 pm, Monday-Friday.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Richemond Dorvil can be reached on (571)-272-7602. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free). If you would like assistance from a USPTO Customer Service Representative or access to the automated information system, call 800-786-9199 (IN USA OR CANADA) or 571-272-1000.

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